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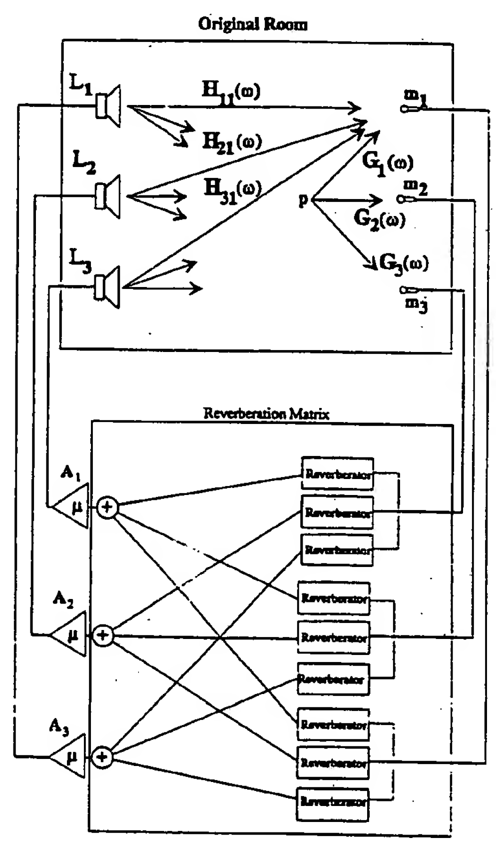
(54) Title: WIDEBAND ASSISTED REVERBERATION SYSTEM

(57) Abstract

A wideband assisted reverberation system has multiple microphones (M1-M3) to pick up reverberant sound in a room, multiple loudspeakers (L1-L3) to broadcast sound into the room, and a reverberation matrix connecting a similar bandwidth signal from the microphones (m) through reverberators to the loudspeakers (L). Preferably the reverberation matrix connects each microphone (m) through one or more reverberators to at least two loudspeakers (L) with cross-linking so that each loudspeaker (L) receives a signal comprising a sum of at least two reverberated microphone signals. Most preferably there is full cross-linking so that every microphone (m) through reverberators to every loudspeaker (L), so that each loudspeaker (L) receives a signal comprising a sum of reverberated microphone signals from every microphone (m).



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WIDEBAND ASSISTED REVERBERATION SYSTEM

TECHNICAL FIELD

The invention relates to assisted reverberation systems. An assisted reverberation system is used to improve and control the acoustics of a concert hall or auditorium.

BACKGROUND ART

There are two fundamental types of assisted reverberation systems. The first is the In-Line System, in which the *direct* sound produced on stage by the performer(s) is picked up by one or more microphones, processed by feeding it through delays, filters and reverberators, and broadcast into the auditorium from several loudspeakers which may be at the front of the hall or distributed around the walls and ceiling. In an In-Line system acoustic feedback (via the auditorium) between the loudspeakers and microphones is not required for the system to work (hence the term in-line).

The second type of assisted reverberation system is the Non-In-Line system, in which a number of microphones pick up the *reverberant* sound in the auditorium and broadcast it back into the auditorium via filters, amplifiers and loudspeakers (and in some variants of the system, via delays and reverberators - see below). The rebroadcast sound is added to the original sound in the auditorium, and the resulting sound is again picked up by the

microphones and rebroadcast, and so on. The Non-In-Line system thus relies on the acoustic feedback between the loudspeakers and microphones for its operation (hence the term non-in-line).

In turn, there are two basic types of Non-In-Line assisted reverberation system. The first is a narrowband system, where the filter between the microphone and loudspeaker has a narrow bandwidth. This means that the channel is only assisting the reverberation in the auditorium over the narrow frequency range within the filter bandwidth. An example of a narrowband system is the Assisted Resonance system, developed by Parkin and Morgan and used in the Royal Festival Hall in London - see *"Assisted Resonance in the Royal Festival Hall."*, J. Acoust. Soc. Amer, vol 48, pp 1025-1035, 1970. The advantage of such a system is that the loop gain may be relatively high without causing difficulties due to instability. A disadvantage is that a separate channel is required for each frequency range where assistance is required.

The second form of Non-In-Line assisted reverberation system is the wideband system, where each channel has an operating frequency range which covers all or most of the audio range. In such a system the loop gains must be low, because the stability of a wideband system with high loop gains is difficult to maintain. An example of such a system is the Philips MCR ('Multiple Channel amplification of Reverberation') system, which is installed in several concert halls around the world, such as the POC Congress

Centre in Eindhoven - see de Koning S.H., *"The MCR System - Multiple Channel Amplification of Reverberation"*, Phillips Tech. Rev., vol 41, pp 12-23, 1983/4.

There are several variants on the non-in-line systems described above. The Yamaha Assisted Acoustics System (AAS) is a combination in-line/non-in-line system. The non-in-line part consists of a small number of channels, each of which contains a finite impulse response (FIR) filter. This filter provides additional delayed versions of the microphone signal to be broadcast into the room, and is supposedly designed to smooth out the frequency response by placing additional peaks between the original peaks - see F. Kawakami and Y. Shimizu, *"Active Field Control in Auditoria"*, Applied Acoustics, vol 31, pp 47-75, 1990. If this is accomplished then the loop gain may be kept quite high without causing undue colouration, and consequently the number of channels required for a reasonable increase in reverberation time is low. However, the design of the FIR filter is critical: the room transfer functions from each loudspeaker to each microphone must be measured and all FIR filters designed to match them. The FIR filter design can not be carried out individually since each filter affects the room response and hence the required response of the other FIR filters. Furthermore, the passive room transfer functions alter with room temperature, positioning of furniture and occupancy, and so the system must be made adaptive: ie the room transfer functions must be continually measured and the FIR filters

updated at a reasonable rate. The system designers have acknowledged that there is currently no method of designing the FIR filters, and so the system cannot operate as it is intended to.

The in-line part of the AAS system consists of a number of microphones that pick up the direct sound, add a number of short echoes, and broadcast it via separate speakers. The in-line part of the AAS system is designed to control the early reflection sequence of the hall, which is important in defining the quality of the acoustics in the hall. An in-line system could easily be added to any existing non-in-line system to allow control of the early reflection sequence in the same way.

A simple variant on the non-in-line system was described by Jones and Fowweather, *"Reverberation Reinforcement - An Electro Acoustic System for Increasing the Reverberation Time of an Auditorium"*, Acoustica, vol 31, pp 357-363, 1972. They improved the sound of the Renold Theatre in Manchester by picking up the sound transmitted from the hall into the space between the suspended ceiling and the roof with several microphones and broadcasting it back into the chamber. This system is a simple example of the use of a secondary acoustically coupled "room" in a feedback loop around a main auditorium for reverberation assistance.

DISCLOSURE OF INVENTION

The present invention provides an improved or at least alternative form of reverberation system.

In its simplest form in broad terms the invention comprises a wideband assisted reverberation system, comprising:

multiple microphones to pick up reverberant sound in a room,

multiple loudspeakers to broadcast sound into the room, and

a diagonal reveration matrix connecting a similar bandwidth signal from each microphone through a reverberator to a loudspeaker.

Preferably the reverberation matrix connects a similar bandwidth signal from each microphone through one or more reverberators to two or more separate loudspeakers, each of which receives a signal comprising one reverberated microphone signal.

More preferably the reverberation matrix connects a similar bandwidth signal from each microphone through one or more reverberators per microphone to one or more loudspeakers, each of

which receives a signal comprising a sum of one or more reverberated microphone signals.

Very preferably the reverberation matrix connects a similar bandwidth signal from each microphone through one or more reverberators to at least two loudspeakers each of which receives a signal comprising a sum of at least two reverberated microphone signals.

Most preferably the reverberation matrix connects a similar bandwidth signal from every microphone through one or more reverberators to every loudspeaker, each of which receives a signal comprising a sum of reverberated microphone signals from every microphone.

In any of the above cases the reverberation matrix may connect at least eight microphones to at least eight loud speakers, or groups of at least eight microphones to groups of at least eight loudspeakers.

A maximum of NK crosslinks between microphones and loudspeakers is achievable where N is the number of microphones and K the number of loud speakers, but it is possible that there are less than NK crosslink connections between the microphones and loudspeakers, provided that the output from at least one microphone

is passed through at least two reverberators and the output of each reverberator is connected to a separate loudspeaker.

The system of the invention simulates placing a secondary room in a feedback loop around the main auditorium with no two-way acoustic coupling. The system of the invention allows the reverberation time in the room to be controlled independently of the steady state energy density by altering the apparent room volume.

BRIEF DESCRIPTION OF DRAWINGS

The invention will now be further described with reference to the accompanying drawings, by way of example and without intending to be limiting. In the drawings:

Fig. 1 shows a typical prior art wide band non-in-line assisted reverberation system,

Fig. 2 shows a wide band non-in-line system of the invention,

Fig. 3 is a block diagram of a simplified assisted reverberation transfer function for low loop gains, and

Fig. 4 shows a preferred form multi input, multi output N channel reverberator design of the invention.

DESCRIPTION OF PREFERRED FORMS

Fig. 1 shows a typical prior art wideband, N microphone, K loudspeaker, non-in-line assisted reverberation system (with N=K=3 for simplicity of the diagram). Each of microphones m_1, m_2 and m_3 picks up the reverberant sound in the auditorium and sends it via one of filters f_1, f_2 and f_3 and amplifiers A_1, A_2 and A_3 of gain μ to a respective single loudspeaker L_1, L_2 and L_3. In an MCR system the filters are used to tailor the loop gain as a function of frequency to get a reverberation time that varies slowly with frequency - they have no other appreciable effect on the system behaviour. In the Yamaha system the filters contain an additional FIR filter which provides extra discrete echoes, and whose responses are in theory chosen to minimise peaks in the overall response and allow higher loop gains, as discussed above. The filter block in both MCR and Yamaha systems may also contain extra processing to adjust the loop gain to avoid instability, and switching circuitry for testing and monitoring.

Fig. 2 shows a wideband, N microphone, K loudspeaker non-in-line system of the invention. Each of microphones m_1, m_2 and m_3 picks up the reverberant sound in the auditorium. Each microphone signal is split into a number K of separate paths, and each 'copy' of the microphone signal is transmitted through a reverberator, (the reverberators typically have a similar reverberation time but may have a different reverberation time). Each microphone signal is connected to each of K loudspeakers

through the reverberators, with the output of one reverberator from each microphone being connected to each of the amplifiers A_1 to A_3 and to loudspeakers L_1 to L_3 as shown i.e. one reverberator signal from each microphone is connected to each loudspeaker and each loudspeaker has connected to it the signal from each microphone, through a reverberator. In total there are NK connections between the microphones and the loudspeakers.

The system of reverberators may be termed a 'reverberation matrix'. It simulates a secondary room placed in a feedback loop around the main auditorium. It can most easily be implemented using digital technology, but alternative electroacoustic technology, such as a reverberation plate with multiple inputs and outputs, may also be used.

While in Fig. 2 each microphone signal is split into K separate paths through K reverberators resulting in NK connections to K amplifiers and loudspeakers, the microphone signals could be split into less than K paths and coupled over less than K reverberators i.e. each loudspeaker may have connected to it the signal from at least two microphones each through a reverberator, but be cross-linked with less than the total number of microphones. For example, in the system of Fig. 2 the reverberation matrix may split the signal from each of microphones m_1, m_2 and m_3 to feed two reverberators instead of three, and the reverberator output from microphone m_1 may then be connected to speakers L_1 and L_3, from

microphone m_2 to speakers L_1 and L_2, and from microphone m_3 to speakers L_2 and L_3.

It can be shown that the system performance is governed by the minimum of N and K, and so systems of the invention where N=K are preferred.

In Fig. 2 each loudspeaker indicated by L_1, L_2 and L_3 could in fact consist of a group of two or more loudspeakers positioned around an auditorium.

In Fig. 2 the signal from the microphones is split prior to the reverberators but the same system can be implemented by passing the supply from each microphone through a single reverberator per microphone and then splitting the reverberated microphone signal to the loudspeakers.

Fig. 3 shows a system with three microphones, three loudspeakers, and three groups of three reverberators but as stated other arrangements are possible, of a single or two microphones, or four or five or more microphones, feeding one or two, or four or five or more loudspeakers or groups of loudspeakers, through one or two, or four or five or more groups of one, two, four or five or more reverberators for example.

The system of the invention may be used in combination with or be supplemented by any other assisted reverberation system such as an in-line system for example. An in-line system may be added to allow control of the early reflection sequence for example.

Very preferably the reverberators produce an impulse response consisting of a number of echoes, with the density of echoes increasing with time. The response is typically perceived as a number of discernible discrete early echoes followed by a large number of echoes that are not perceived individually, rather they are perceived as 'reverberation'. Reverberators typically have an infinite impulse response, and the transfer function contains poles and zeros. It is however possible to produce a reverberator with a finite impulse response and a transfer function that contains only zeros. Such a reverberator would have a truncated impulse response that is zero after a certain time. The criterion that a reverberator must meet is the high density of echoes that are perceived as room reverberation.

Each element in the reverberation matrix may be denoted $X_{nk}(\omega)$ (the transfer function from the nth microphone to the kth loudspeaker). The system analysis is described in terms of an NxK matrix of the $X_{nk}(\omega)$ and a KxN matrix of the original room transfer functions between the kth loudspeaker and the nth microphone,

denoted $H_{kn}(\omega)$. This analysis produces a vector equation for the transfer functions;

$$\tilde{Y}(\omega) = [Y_1(\omega), Y_2(\omega), \ldots, Y_N(\omega)]^T \tag{1}$$

from a point in the original auditorium to each microphone as follows;

$$\tilde{Y}(\omega) = \frac{1}{V_0(\omega)} \tilde{V}(\omega) = [\tilde{I} - \mu \tilde{H}^T(\omega) \tilde{X}^T(\omega)]^{-1} \tilde{G}(\omega) \tag{2}$$

where $V_0(\omega)$ is the spectrum of the excitation signal input to a speaker at a point p in the room,

$$\tilde{V}(\omega) = [V_1(\omega), V_2(\omega), \ldots, V_N(\omega)]^T, \tag{3}$$

is a vector containing the spectra at each microphone with the system operating,

$$\tilde{G}(\omega) = [G_1(\omega), G_2(\omega), \ldots, G_N(\omega)]^T, \tag{4}$$

is a vector of the original transfer functions from p to each microphone with the system off,

$$\tilde{X}(\omega)=\begin{bmatrix} X_{11}(\omega) & X_{12}(\omega) & X_{13}(\omega) & \dots & X_{1K}(\omega) \\ X_{21}(\omega) & X_{22}(\omega) & X_{23}(\omega) & \dots & X_{2K}(\omega) \\ X_{31}(\omega) & X_{32}(\omega) & X_{33}(\omega) & \dots & X_{3K}(\omega) \\ \cdot & & & & \cdot \\ \cdot & & & & \cdot \\ X_{N1}(\omega) & X_{N2}(\omega) & X_{N3}(\omega) & \cdot & X_{NK}(\omega) \end{bmatrix}, \tag{5}$$

is the matrix of reverberators, and

$$\tilde{H}(\omega)=\begin{bmatrix} H_{11}(\omega) & H_{12}(\omega) & H_{13}(\omega) & \dots & H_{1N}(\omega) \\ H_{21}(\omega) & H_{22}(\omega) & H_{23}(\omega) & \dots & H_{2N}(\omega) \\ H_{31}(\omega) & H_{32}(\omega) & H_{33}(\omega) & \dots & H_{3N}(\omega) \\ \cdot & & & & \cdot \\ \cdot & & & & \cdot \\ \cdot & & & & \cdot \\ H_{K1}(\omega) & H_{K2}(\omega) & H_{K3}(\omega) & \dots & H_{KN}(\omega) \end{bmatrix} \tag{6}$$

is the matrix of original transfer functions, $H_{kn}(\omega)$ from the kth loudspeaker to the nth microphone with the system off.

With the transfer functions to the system microphones derived, the general response to any other M receiver microphones in the room may be written as

$$\tilde{Z}(\omega)=\frac{1}{V_0(\omega)}[\tilde{E}(\omega)+\mu\tilde{F}^T(\omega)\,\tilde{X}^T(\omega)\,[\tilde{I}-\mu\tilde{H}^T(\omega)\,\tilde{X}^T(\omega)]^{-1}\tilde{G}(\omega)] \tag{7}$$

where

$$\tilde{E}(\omega)=[E_1(\omega),E_2(\omega),\dots,E_M(\omega)]^T, \tag{8}$$

is the original vector of transfer functions to the M receiver microphones in the room and

$$\tilde{F}(\omega) = \begin{bmatrix} F_{11}(\omega) & F_{12}(\omega) & F_{13}(\omega) & \dots & F_{1M}(\omega) \\ F_{21}(\omega) & F_{22}(\omega) & F_{23}(\omega) & \dots & F_{2M}(\omega) \\ F_{31}(\omega) & F_{32}(\omega) & F_{33}(\omega) & \dots & F_{3M}(\omega) \\ \cdot & & & & \cdot \\ \cdot & & & & \cdot \\ \cdot & & & & \cdot \\ F_{K1}(\omega) & F_{K2}(\omega) & F_{K3}(\omega) & \dots & F_{KM}(\omega) \end{bmatrix} \quad (9)$$

is another matrix of room transfer functions from the K loudspeakers to the M receiver microphones.

To determine the steady state energy density level of the system for a constant input power, a power analysis of the system may be carried out assuming that each $E_n(\omega)$, $G_n(\omega)$, $X_{nk}(\omega)$, $H_{kn}(\omega)$ and $F_{km}(\omega)$ has unity mean power gain and a flat locally averaged response. The mean power of the assisted system for an input power P is then given by

$$P_{ass} = \frac{P}{1-\mu^2 KN} \tag{10}$$

Since the power is proportional to the steady state energy density which is inversely proportional to the absorption, the absorption is reduced by a factor $(1-\mu^2KN)$. The reverberation time of a room is given approximately by

$$T = .16\frac{V}{A} \tag{11}$$

where V equals the apparent room volume and A equals the apparent room absorption. Hence the change in absorption also increases the reverberation time by $1/(1-\mu^2KN)$. The MCR system has no cross coupling and produces a power and reverberation time increase of $1/(1-\mu^2N)$. The two systems produce the same energy density boost

and reverberation time with similar colouration if the MCR system loop gain μ is increased by a factor $\sqrt{K}$.

The reverberation time of the assisted system is increased when the apparent room absorption is decreased. It is *also* increased if the apparent room volume is increased, from equation 1 1. The solution in equation 7 may be written as

$$\tilde{Y}(\omega) = \tilde{E}(\omega) + \frac{\mu \tilde{F}^T(\omega)\tilde{X}^T(\omega)}{\det[\bar{I} - \mu \tilde{H}^T(\omega)\tilde{X}^T(\omega)]} Adj[\bar{I} - \mu \tilde{H}^T(\omega)\tilde{X}^T(\omega)]\tilde{G}(\omega) \qquad (12)$$

where det is the determinant of the matrix and Adj denotes the adjoint matrix.

For low loop gains the transfer function from a point in the room to the ith receiver microphone may be simplified by ignoring all squared and higher powers of μ, and all μ terms in the adjoint;

$$Y_i(\omega) \approx E_i(\omega) + \frac{\mu \sum_{l=1}^{N} \sum_{k=1}^{K} G_l(\omega) X_{lk}(\omega) F_{ki}(\omega)}{1 - \mu \sum_{n=1}^{N} \sum_{k=1}^{K} X_{nk}(\omega) H_{kn}(\omega)} \qquad (13)$$

Equation 13 reveals that the assisted system may be modelled as a sum of the original transfer function, $E_i(\omega)$, plus an additional transfer function consisting of the responses from the lth system microphone to the ith receiver microphone in series with

a recursive feedback network, as shown in figure 3. The overall reverberation time may thus be increased by altering the reverberation time of the recursive network. This may be done by increasing μ, which also alters the absorption, or independently of the absorption by altering the phase of the $X_{nk}(\omega)$ (This also increases the reverberation time of the feedforward section). The recursive filter resembles a simple comb filter, but has a more complicated feedback network than that of a pure delay. The reverberation time of a comb filter with delay τ and gain μ is equal to $-3\tau/\log(\mu)$. T_{rec} may therefore be defined as;

$$T_{rec} = \frac{3\frac{\int \phi'_{rec}(\omega)\, M_{rec}^{2}(\omega)\, d\omega}{\int M_{rec}^{2}(\omega)\, d\omega}}{\log[\mu \overline{M}_{rec}]} \qquad (14)$$

where $M_{rec}(\omega)$ is the overall magnitude (with mean $\overline{M}_{rec}$) and $-\phi_{rec}'(\omega)$ is the overall group delay of the feedback network. Thus the reverberation time, and hence the volume, may be independently controlled by altering the phase of the reverberators, $X_{nk}(\omega)$. This feature is not available in previous systems which either have no reverberators in the feedback loop as in the Philips MCR system - or which have a fixed acoustic room in the feedback loop which is not easily controlled. The Yamaha system will produce a limited change in apparent volume, but this cannot be arbitrarily altered since a) the FIR filters have a finite number of echoes which cannot be made arbitrarily long without producing unnaturalness

such as flutter echoes (see Kawakimi and Shimuzu above), and b) the FIR filters also have to maintain stability at high loop gains and so their structure is constrained. The matrix of feedback reverberators introduced here has a considerably higher echo density so that flutter echoes problems are eliminated, and the fine structure of the reverberators has no bearing on the colouration of the system since the matrix is intended to be used in a system with a reasonably large number of microphones and loudspeakers and low loop gains. The reverberation matrix thus allows independent control of the apparent volume of the assisted auditorium without altering the perceived colouration by altering the reverberation time of the matrix without altering its mean gain.

Fig. 4 shows one possible implementation of an N channel input, N channel output reverberator. The N inputs I_1, to I_N are cross coupled through an N by N gain matrix and the outputs are connected to N delay lines. The delay line outputs O_1 to O_N are fed back and summed with the inputs. It can be shown that the system is unconditionally stable if the gain matrix is equal to an orthonormal matrix scaled by a gain μ which is less than one.

The foregoing describes the invention including preferred forms thereof. Alterations and modifications as will be obvious to those skilled in the art are intended to be incorporated in the scope thereof as defined in the claims.

CLAIMS

1. A wideband assisted reverberation system, including:

multiple microphones to pick up reverberant sound in a room,

multiple loudspeakers to broadcast sound into the room, and

a diagonal reverberation matrix connecting a similar bandwidth signal from each microphone through a reverberator to a loudspeaker.

2. A wideband assisted reverberation system, including:

multiple microphones to pick up reverberant sound in a room,

multiple loudspeakers to broadcast sound into the room, and

a reverberation matrix connecting a similar bandwidth signal from each microphone through one or more reverberators to two or more separate loudspeakers and each of which receives a signal comprising one reverberated microphone signal.

3. A wideband non-in-line assisted reverberation system, including:

multiple microphones to pick up reverberant sound in a room,

multiple microphones to broadcast sound into the room, and

a reverberation matrix connecting a similar bandwidth signal from each microphone through one or more reverberators per microphone to one or more loudspeakers, each of which receives a signal comprising a sum of one or more reverberated microphone signals.

4. A wideband assisted reverberation system as claimed in claim 3, wherein the reverberation matrix connects a similar bandwidth signal from each microphone through one or more reverberators to at least two loudspeakers each of which receives a signal comprising a sum of at least two reverberated microphone signals.

5. A wideband assisted reverberation system as claimed in claim 3, wherein the reverberation matrix connects a similar bandwidth signal from every microphone through one or more reverberators to every loudspeaker, each of which receives a signal

comprising a sum of reverberated microphone signals from every microphone.

6. A wideband assisted reverberation system as claimed in any one of claims 3 to 5, wherein the reverberation matrix connects at least eight microphones to at least eight loud speakers, or where groups of at least eight microphones are connected to groups of at least eight loudspeakers.

7. A wideband assisted reverberation system as claimed in any one of the preceding claims, having impulse responses consisting of multiple echoes of increasing density with time.

8. A diagonal reverberation matrix comprising multiple microphone inputs and multiple reverberation outputs and connecting a similar bandwidth signal from each microphone input through a reverberator to a reverberation output.

9. A reverberation matrix comprising multiple microphone inputs and multiple reverberation outputs and connecting a similar bandwidth signal from each input through one or more reverberators per microphone input to a separate reverberation output.

10. A reverberation matrix as claimed in claim 9, wherein the reverberation matrix connects a similar bandwidth signal from each microphone input through reverberators to one or more outputs with

each reverberator output signal comprising a sum of one or more reverberated input signals.

11. A reverberation matrix as claimed in claim 9, wherein the reverberation matrix connects a similar bandwidth signal from each microphone input through one or more reverberators to at least two reverberation outputs with each reverberator output signal comprising a sum of at least two reverberated microphone input signals.

12. A reverberation matrix as claimed in claim 9, wherein the reverberation matrix connects a similar bandwidth signal from every microphone input through reverberators to every reverberator output.

13. A reverberation matrix as claimed in any one of claims 9 to 12, wherein the reverberation matrix connects at least eight microphone inputs to at least eight reverberator outputs, or groups of at least eight microphone inputs to groups of at least eight reverberator outputs.

14. A reverberation matrix as claimed in any one of claims 9 to 13, having impulse responses from any input to any output consisting of multiple echoes of increasing density with time.

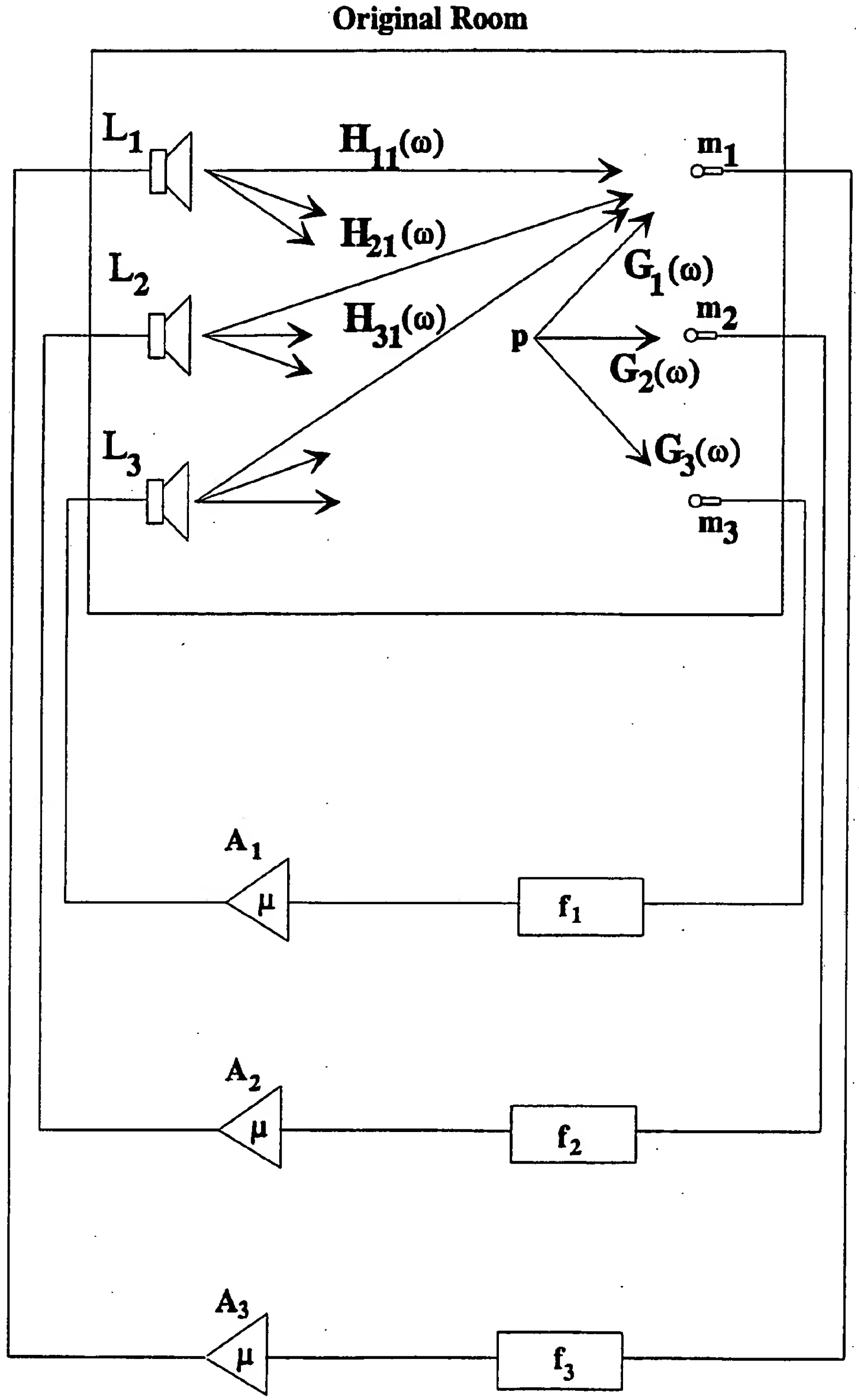
Original Room
L_1
$H_{11}(\omega)$
m_1
$H_{21}(\omega)$
L_2
$G_1(\omega)$
$H_{31}(\omega)$
m_2
p
$G_2(\omega)$
L_3
$G_3(\omega)$
m_3
A_1
μ
f_1
A_2
μ
f_2
A_3
μ
f_3

Figure 1

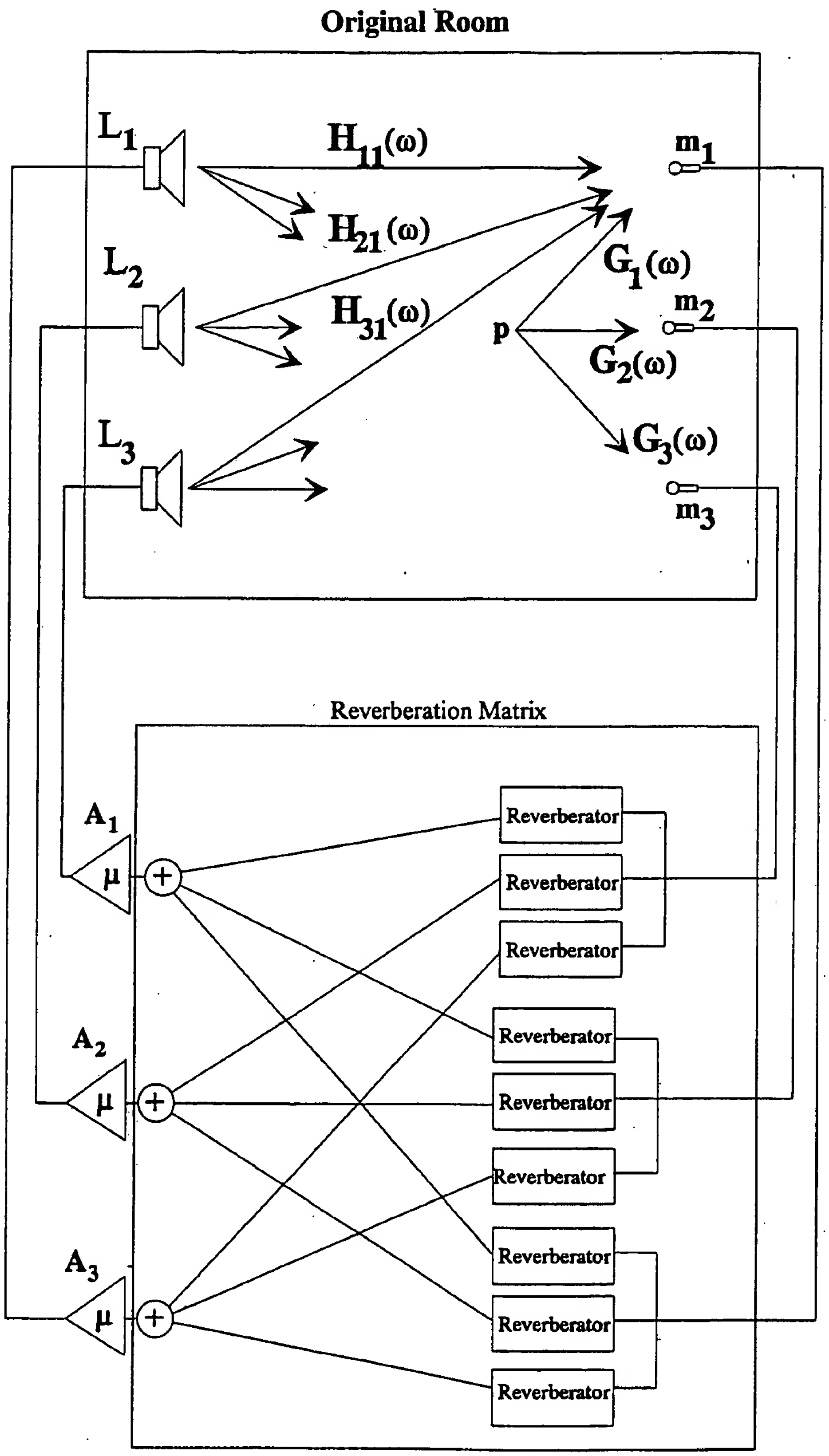
Original Room
L_1
L_2
L_3
$H_{11}(\omega)$
$H_{21}(\omega)$
$H_{31}(\omega)$
m_1
m_2
m_3
p
$G_1(\omega)$
$G_2(\omega)$
$G_3(\omega)$
Reverberation Matrix
A_1
A_2
A_3
μ
Reverberator

Figure 2

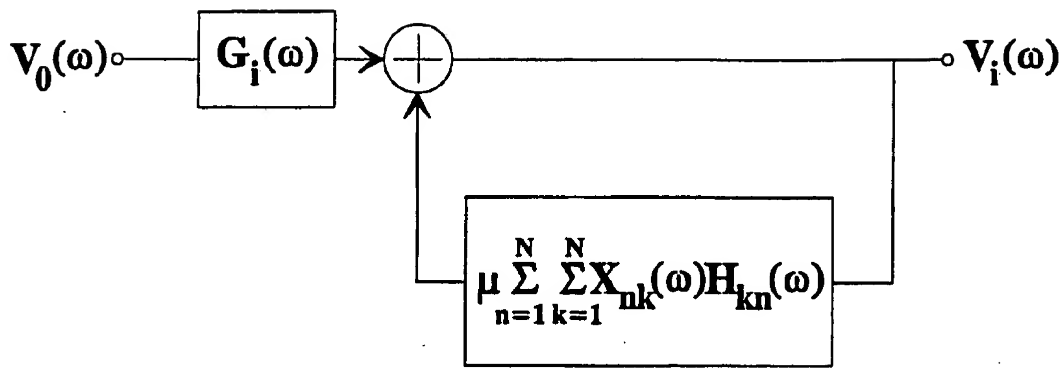
$V_0(\omega)$
$G_i(\omega)$
$V_i(\omega)$
$\mu \sum_{n=1}^{N} \sum_{k=1}^{N} X_{nk}(\omega) H_{kn}(\omega)$

Figure 3

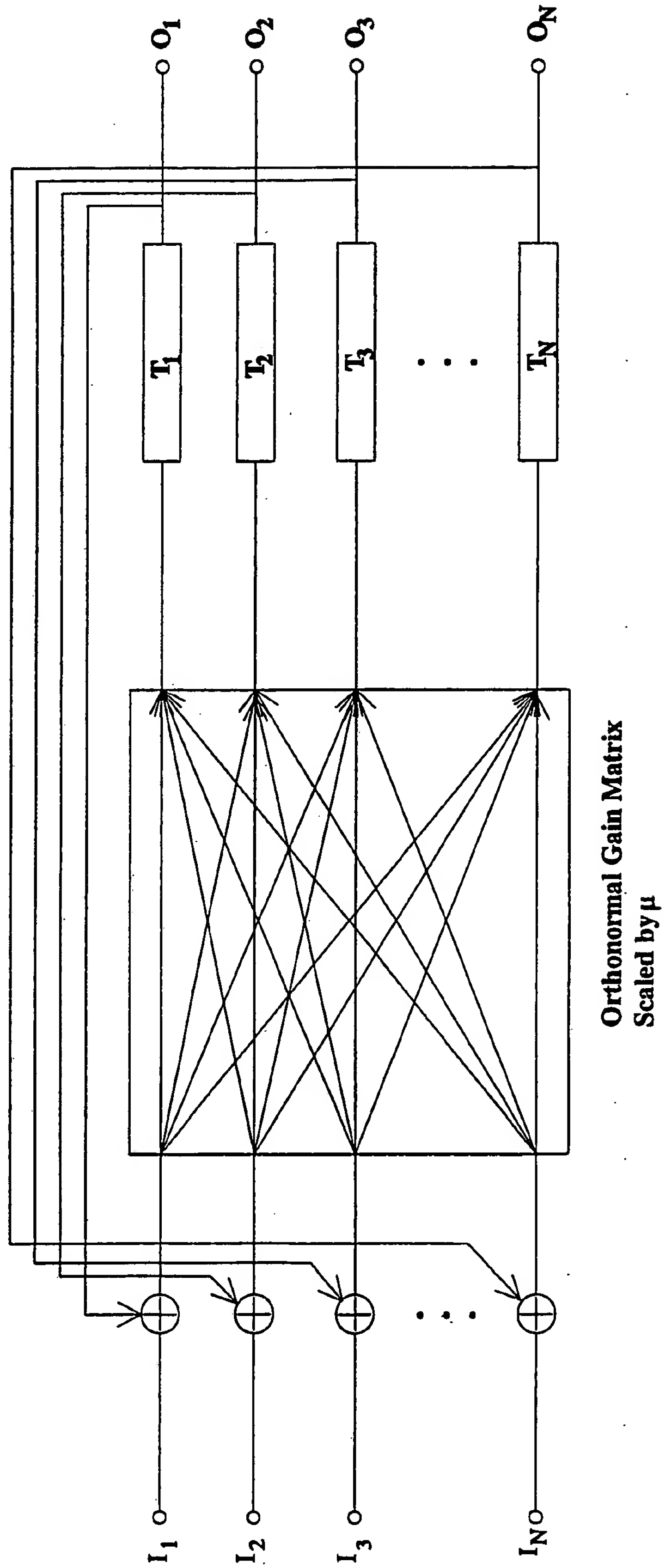
O1
O2
O3
ON
T1
T2
T3
TN
Orthonormal Gain Matrix
Scaled by μ
I1
I2
I3
IN

Figure 4

INTERNATIONAL SEARCH REPORT

International Application No PCT/NZ 93/00041

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)[6]

According to International Patent Classification (IPC) or to both National Classification and IPC

Int.Cl. 5 G10K15/08

II. FIELDS SEARCHED

Minimum Documentation Searched[7]

Classification System	Classification Symbols
Int.Cl. 5	G10K ; G10H

Documentation Searched other than Minimum Documentation
to the Extent that such Documents are Included in the Fields Searched[8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category °	Citation of Document, [11] with indication, where appropriate, of the relevant passages [12]	Relevant to Claim No.[13]
A	US,A,5 109 419 (GRIESINGER) 28 April 1992 see column 5, line 4 - column 5, line 54; figure 4 ---	1
A	DE,A,4 022 217 (PIONEER ELECTRONIC CORP.) 6 June 1991 see column 3, line 13 - column 4, line 46; figure 1 ---	1
A	EP,A,0 335 468 (BIRCH WOOD ACOUSTICS NEDERLAND BV) 4 October 1989 see page 8, line 34 - page 8, line 50; figures 6,7 -----	1

° Special categories of cited documents : [10]

"A" document defining the general state of the art which is not considered to be of particular relevance

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"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
26 AUGUST 1993	29. 09. 93
International Searching Authority	**Signature of Authorized Officer**
EUROPEAN PATENT OFFICE	ANDERSON A.TH.

1

ANNEX TO THE INTERNATIONAL SEARCH REPORT ON INTERNATIONAL PATENT APPLICATION NO.

NZ 9300041
SA 74420

This annex lists the patent family members relating to the patent documents cited in the above-mentioned international search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information. 26/08/93

Patent document cited in search report	Publication date	Patent family member(s)		Publication date
US-A-5109419	28-04-92	None		
DE-A-4022217	06-06-91	JP-A-	3171900	25-07-91
		US-A-	5119420	02-06-92
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		JP-T-	2503721	01-11-90
		WO-A-	8909465	05-10-89
		US-A-	5142586	25-08-92

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82